DEFENSE INFORMATION SYSTEMS AGENCY
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Joint Interoperability Test Command (JTE)   **12 Mar 09**

MEMORANDUM FOR DISTRIBUTION

SUBJECT: Extension of the Special Interoperability Test Certification of the Avaya S8400
Digital Switching System with Software Release Communication Manager (CM)
4.0 (R014x.00.2.731.7: Super Patch 14419)

References: (a) DoD Directive 4630.5, "Interoperability and Supportability of Information
Technology (IT) and National Security Systems (NSS)," 5 May 2004
(b) CJCSI 6212.01D, "Interoperability and Supportability of Information
Technology and National Security Systems," 8 March 2006
(c) through (f), see Enclosure

1. References (a) and (b) establish the Defense Information Systems Agency (DISA), Joint
Interoperability Test Command (JITC), as the responsible organization for interoperability test
certification.

2. The Avaya S8400 Digital Switching System with Software Release CM 4.0
(R014x.00.2.731.7: Super Patch 14419) is hereinafter referred to as the System Under Test
(SUT). The SUT met all of its critical interoperability requirements and is certified as
interoperable for joint use within the Defense Switched Network (DSN). The SUT was tested
and met the critical interoperability requirements for the following DSN switch types: Private
Branch Exchange (PBX) 1, PBX 2, and Deployable Voice Exchange (DVX). The SUT is
certified to support DSN Assured Services over Internet Protocol with any Assured Services
Voice Application Local Area Network (ASVALAN) on the Unified Capabilities (UC)
Approved Products List (APL). The SUT is also certified for joint use with any Voice
Application Local Area Network (VALAN) on the UC APL. However, since VALANs do not
support the Assured Services Requirements detailed in reference (c), Command and Control (C2)
users and Special C2 users are not authorized to be served by the SUT connected to a VALAN.

The S8400 series media servers work in conjunction with the G650 complementary media
gateways which support multi-protocol environments for concurrent support of Time Division
Multiplex (TDM) and Internet Protocol (IP)-based telephony. The SUT is capable of supporting
a maximum of five G650s from one port network. JITC, however, conducted testing on the SUT
using only one G650. Based on this testing and through analysis, this certification applies to
SUTs that are configured with up to five G650s. The SUT offers an internal Automated Call
Distributor (ACD). The ACD was tested and is covered under this certification. The SUT does
not offer an internal voicemail capability; however, the SUT is certified for external voicemail
through the use of the 2-wire digital proprietary interface. The SUT is certified for conferencing
through use of any external conferencing bridge that is on the UC APL. The identified test

JITC Memo, JTE, Extension of the Special Interoperability Test Certification of the Avaya S8400 Digital Switching System with Software Release Communication Manager (CM) 4.0 (R014x.00.2.731.7: Super Patch 14419)

discrepancies shown in the SUT Interoperability Test Summary, which remained open after Super Patch 14419 was applied and regression tested, have an overall minor operational impact. No other configurations, features, or functions, except those cited within this report, are certified by the JITC or authorized by the Program Management Office for use within the DSN. This certification expires upon changes that affect interoperability, but no later than three years from the date of the original memorandum (30 October 2007).

3. The extension of this certification is based upon a desktop review. The original certification is based on interoperability testing conducted by JITC and a review of the vendor's Letters of Compliance (LoC). Interoperability testing was conducted by JITC's Global Information Grid Network Test Facility at Fort Huachuca, Arizona, from 4 June through 13 July 2007 and documented in reference (d). Regression testing was conducted from 7 through 10 August 2007. Review of the LoC was completed on 13 August 2007. All testing was completed with software release CM 4.0 (R014x.00.2.731.7: Super Patch 14419). A desktop review was requested to include R014x.00.2.732.1: Super Patch 16538. This software load was validated during the Avaya S8300/G350 testing and received APL certification. The desktop review request was approved on 13 February 2009.

4. The interoperability test summary of the SUT is contained in Table 1. The PBX 1 Capability Requirements (CRs) and Feature Requirements (FRs) are listed in Table 2. If a switch meets the PBX 1 requirements, it meets the lesser requirements of a PBX 2. The comparison between PBX 1 and DVX requirements and interoperability summary is listed in Table 3. This interoperability test status is based on the SUT's ability to meet:

   a. DSN services for Network and Applications specified in reference (c).

   b. PBX 1 and DVX interface and signaling requirements for trunks/lines specified in reference (e) verified through JITC testing and/or vendor submission of LoC.

   c. PBX 1 and DVX CRs/FRs specified in reference (e) verified through JITC testing and/or vendor submission of LoC.

   d. Internet Protocol version 6 requirements specified in reference (e), paragraph 1.7, Table 1-4, verified through vendor submission of LoC signed by the Vice President of the company.

   e. The overall system interoperability performance derived from test procedures listed in reference (f).

JITC Memo, JTE, Extension of the Special Interoperability Test Certification of the Avaya S8400 Digital Switching System with Software Release Communication Manager (CM) 4.0 (R014x.00.2.731.7: Super Patch 14419)

Table 1. SUT Interoperability Test Summary

DSN Trunk Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
T1 CAS (DTMF, DP, MFR1)	No	Certified	Met all CRs and FRs with the following minor exceptions: The SUT fails to remove a yellow alarm condition after a DS1 has been broken and restored within GSCR specification.[1] The SUT T1 CAS preemption signal generation is out of tolerance.[2] The SUT recognizes E1 and T1 CAS wink start signals greater than the maximum interval as valid.[3] During a remote busy condition on a T1 CAS or E1 CAS, the SUT takes approximately 5 minutes to change the status of the timeslots from an "In-Service/Active" state to a "Far-End-Busy" state.[4]
E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Certified	Met all CRs and FRs with the following minor exceptions: The SUT fails to remove a yellow alarm condition after a DS1 has been broken and restored within GSCR specification.[1] The SUT recognizes E1 and T1 CAS wink start signals greater than the maximum interval as valid.[3] During a remote busy condition on a T1 CAS or E1 CAS, the SUT takes approximately 5 minutes to change the status of the timeslots from an "In-Service/Active" state to a "Far-End-Busy" state.[4]
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Met all CRs and FRs with the following minor exceptions: The SUT fails to remove a yellow alarm condition after a DS1 has been broken and restored within GSCR specification.[1] Failure to maintain busy out condition after restart messages are received from the distant switch.[5]
E1 ISDN PRI	No	Not Tested	The SUT offers an E1 ISDN PRI interface; however, this interface was not tested and is not covered under this certification. Since this is not a required interface for a PBX 1 or DVX, there is no operational impact.[6]
DSN Line Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
2-Wire Analog (GR-506-CORE)	Yes	Certified	Met all CRs and FRs with the following minor exceptions: The precedence above ROUTINE ring cadence is not in accordance with GSCR specification.[7] The call pick-up feature does not pick-up the call with the highest precedence or longest ringing call first.[8] Three-way conference members do not maintain their assigned precedence levels.[9]
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Certified	Met all CRs and FRs with the following minor exceptions: The precedence above ROUTINE ring cadence is not in accordance with GSCR specification.[7] The call pick-up feature does not pick-up the call with the highest precedence or longest ringing call first.[8] Three-way conference members do not maintain their assigned precedence levels.[9]
2-Wire Proprietary Digital	No	Certified	Met all CRs and FRs with the following minor exceptions: The precedence above ROUTINE ring cadence is not in accordance with GSCR specification.[7] The call pick-up feature does not pick-up the call with the highest precedence or longest ringing call first.[8] Three-way conference members do not maintain their assigned precedence levels.[9]
VoIP (IEEE 802.3)	No	Certified	Met all CRs and FRs with the following minor exceptions: The precedence above ROUTINE ring cadence is not in accordance with GSCR specification.[7] The call pick-up feature does not pick-up the call with the highest precedence or longest ringing call first.[8] Three-way conference members do not maintain their assigned precedence levels.[9]
Voicemail			
Interface	**Critical**	**Status**	**Remarks**
2-Wire Proprietary Digital	No	Certified	Met all CRs and FRs.
Automated Call Distributor			
Internal	No	Certified	Met all CRs and FRs.

Table 1. SUT Interoperability Test Summary (continued)

DSN Features and Capabilities			
Features and Capabilities	**Critical**	**Status**	**Remarks**
Common Features	No	Certified	Met all CRs and FRs with the following minor exception: Selective Call Rejection is not supported by the SUT.[10]
Attendant	No	Certified	Met all CRs and FRs with the following minor exception: The SUT attendant console does not support the automatic recall feature.[11]
Public Safety	Yes	Certified	Met all CRs and FRs with the following minor exception: Tandem call trace of a distant office DN is not supported by SUT.[12]
Preset Conferencing	No	Certified	This feature is met through the use of the Compunetx Contex® 240.
Nailed-up Connections	No	Not Tested	This feature is not supported. Since this is not a required feature for a PBX 1 or DVX, there is no operational impact.[13]
DSN Hotline Services	Yes	Certified	The SUT met all CRs and FRs. Hotline Services is required only for analog interfaces. The SUT supports Hotline Services only with analog stations.
ISDN Services (EKTS)	No	Certified	Met all CRs and FRs with the following minor exceptions: When an EKTS member is assigned to a MLHG, a call to that EKTS member fails to ring the other EKTS members.[14] When an intercom call is placed on an EKTS station, the primary DN of the calling EKTS user is used and the station is made busy.[15]
Synchronization	Yes	Certified	Met all CRs and FRs.
Reliability	Yes	Certified	Met all CRs and FRs.
Security	Yes	See note 16.	See note 16.
VoIP			
VoIP System	No	Certified	Met all CRs and FRs. The SUT is certified for VoIP with any VALAN or ASVALAN on the UC APL. See note 17.
Network Gateways			
Gateway	**Interface & Signaling**	**Critical**	**Status**	**Remarks**
PSTN	T1 CAS (DTMF, DP)	No	Certified	Met all CRs and FRs.
	T1 CAS (MFR1)	No	Certified	Met all CRs and FRs.
	E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Certified	Met all CRs and FRs.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all CRs and FRs.
	E1 ISDN PRI	No	Not Tested	The SUT offers an E1 ISDN PRI interface; however, this interface was not tested and is not covered under this certification. Since this is not a required interface for a PBX 1 or DVX, there is no operational impact.[6]
	Ground Start Line	Yes	Certified	Met all CRs and FRs.
DRSN	TPC 2-Wire analog (GR-506-CORE)	Yes	Certified	Met all CRs and FRs. See note 18.

Table 1. SUT Interoperability Test Summary (continued)

NOTES :

1   The SUT fails to remove a yellow alarm condition after a DS1 has been broken and restored within GSCR specification. The requirement states that the yellow alarm should be removed 15 seconds +/- 5 seconds upon DS1 restoration. The SUT removes the yellow alarm 30 seconds after the DS1 is restored. The operational impact is minor.

2   The SUT T1 CAS preemption signal generation is out of tolerance. The preemption signal generated by the SUT was measured 2 ms outside the GSCR required preemption signal of 345 ms +/- 5 ms. The operational impact is minor.

3   The SUT recognizes E1 and T1 CAS wink start signals greater than the maximum interval as valid. The SUT recognizes wink start signals from 100 ms to 395 ms as valid. The GSCR requirement specifies the wink start recognition range to be between 100 ms and 350 ms. The operational impact is minor.

4   During a remote busy condition on a T1 CAS or E1 CAS, the SUT takes approximately 5 minutes to change the status of the timeslots from an "In-Service/Active" state to a "Far-End-Busy" state. During this period of time, a ROUTINE call attempted over this span receives T-120 and a precedence above ROUTINE call receives Blocked Precedence Announcement. After the state is changed, the correct treatment, an Isolated Code Announcement, is provided to all calls attempted over this span. The operational impact is minor.

5   When the SUT initiates a busy-out condition for a T1 PRI, and if the distant switch sends RESTART messages while the SUT has a busy-out condition, the SUT responds with RESTART ACKNOWLEDGEMENT messages; however, the SUT does not retransmit the SERVICE (Out-Of-Service) message for all of the busied channels. The result is that the distant switch idles the channels that the SERVICE (Out-Of-Service) messages were not retransmitted on. This condition can be eliminated by busying both ends. The operational impact is minor.

6   The SUT offers an E1 ISDN PRI interface; however, this interface was not tested and is not covered under this certification. Therefore, this interface is not certified by the JITC or authorized by the Program Management Office for use within the DSN. Since this is not a required interface for a PBX 1 or DVX, there is no operational impact.

7   The precedence above ROUTINE ring cadence is not in accordance with GSCR specification. Since the cadence is different than a ROUTINE ring cadence, the operational impact is minor.

8   The SUT call pickup feature doesn't retrieve the call with the highest precedence first. The SUT retrieves unanswered call pickup group calls above ROUTINE in a random sequence. The GSCR requires that "If a call pickup group has more than one party in an unanswered condition and the unanswered parties are at different precedence levels, a call pickup attempt in that group shall retrieve the highest precedence call first." All unanswered precedence calls above ROUTINE in the pickup group do divert after 15-45 seconds if unanswered and are positively connected to the attendant, night service, or alternate DN. The operational impact is minor.

9   Three-way conference members do not maintain their assigned precedence levels. Since the SUT classmarks the conference members at the highest precedence level, the operational impact is minor.

10 Selective Call Rejection is not supported by the SUT. Since this is not a required feature for a PBX 1 or DVX, there is no operational impact.

11 The SUT attendant console does not support the automatic recall feature. The SUT does permit the attendant console to extend (camp-on) a caller to a busy station. Since this is not a required feature for a PBX 1 or DVX and the SUT provides this for the subscriber as a feature access code, the operational impact is minor.

12 Tandem call trace of a distant office DN is not supported by SUT. Since this is not a required feature for a PBX 1, there is no operational impact. Although it is a requirement for a DVX, the operational impact is minor.

13 This feature is not supported. Since this is not a required feature for a PBX 1 or DVX, there is no operational impact.

14 When an EKTS member is assigned to a MLHG, a call to that EKTS member fails to ring the other EKTS members. When a call is sent to a MLHG pilot number that causes an EKTS member to ring, all members of the EKTS group should have an incoming call appearance. The EKTS feature is certified as standalone and not when assigned as a member of a MLHG. MLHG interaction with EKTS is a conditional requirement; therefore, the operational impact is minor.

15 When an intercom call is placed on an EKTS station, the primary DN of the calling EKTS user is used and the station is made busy. In accordance with the GSCR specification, the EKTS intercom feature should not affect the busy/idle status of any of the DNs of the calling EKTS user. An EKTS station can have additional call appearances added to compensate for this discrepancy. The operational impact is minor.

16 Security is tested by DISA-led Information Assurance test teams and published in a separate report.

17 An IPv6 capable system or product, as defined in the GSCR, paragraph 1.7, shall be capable of receiving, processing, and forwarding IPv6 packets and/or interfacing with other systems and protocols in a manner similar to that of IPv4. IPv6 capability is currently satisfied by a vendor Letter of Compliance signed by the Vice President of their company. The vendor stated, in writing, compliance to the following criteria by 31 December 2008:
    (a) Conformant with IPv6 standards profile contained in the DISR.
    (b) Maintaining interoperability in heterogeneous environments and with IPv4.
    (c) Commitment to upgrade as the IPv6 standard evolves.
    (d) Availability of contractor/vendor IPv6 technical support.

18 Interoperability Certification of the SUT does not constitute DRSN PM's approval for connectivity to the DRSN. It is the user's responsibility to request connectivity approval directly from the PM.

JITC Memo, JTE, Extension of the Special Interoperability Test Certification of the Avaya S8400 Digital Switching System with Software Release Communication Manager (CM) 4.0 (R014x.00.2.731.7: Super Patch 14419)

Table 1. SUT Interoperability Test Summary (continued)

LEGEND:			
802.3u	Standard for carrier sense multiple access with collision detection at 100 Mbps	IPv4	Internet Protocol version 4
ANSI	American National Standards Institute	IPv6	Internet Protocol version 6
APL	Approved Products List	ISDN	Integrated Services Digital Network
ASVALAN	Assured Services Voice Application Local Area Network	IT	Information Technology
		JITC	Joint Interoperability Test Command
BRI	Basic Rate Interface	LSSGR	Local Access and Transport Area (LATA) Switching System Generic Requirements
CAS	Channel Associated Signaling	Mbps	Megabits per second
CRs	Capability Requirements	MFR1	Multi-Frequency Recommendation 1
DISA	Defense Information Systems Agency	MLHG	Multi-Line Hunt Group
DISR	DoD IT Standards Registry	MLPP	Multi-Level Precedence and Preemption
DN	Directory Number	ms	milliseconds
DoD	Department of Defense	NI 1/2	National ISDN Standard 1 or 2
DP	Dial Pulse	PBX 1	Private Branch Exchange 1
DRSN	Defense Red Switch Network	PM	Program Manager
DSN	Defense Switched Network	PRI	Primary Rate Interface
DS1	Digital Signal Level 1	PSTN	Public Switched Telephone Network
DSS1	Digital Subscriber Signaling 1	SS7	Signaling System 7
DTMF	Dual Tone Multi-Frequency	SUT	System Under Test
DVX	Deployable Voice Exchange	T1	Digital Transmission Link Level 1 (1.544 Mbps)
E1	European Basic Multiplex Rate (2.048 Mbps)	T1.607	ISDN – Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
EKTS	Electronic Key Telephone System		
FRs	Feature Requirements	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
GR	Generic Requirement	TPC	Twisted Pair Copper
GR-506-CORE	LSSGR: Signaling for Analog Interfaces	UC	Unified Capabilities
GSCR	Generic Switching Center Requirements	VALAN	Voice Application Local Area Network
IEEE	Institute of Electrical and Electronics Engineers	VoIP	Voice over Internet Protocol

Table 2. PBX 1 Requirements

DSN Trunk Interfaces				
Interface	**Critical**	**Requirements Required or Conditional**		**References**
T1 CAS (MFR1, DTMF, DP) E1 CAS (MFR1, DTMF, DP)	No No (Europe only)	Trunking	• Framing (R) • Line Code (R) • Signaling (R) • Alarm and Restoral Requirements (R) • Alarm and Restoral Requirements (C) • WWNDP (R) • Outpulsing digit formats (C: CAS only) • Routing (C) • Trunk Groups (C) • CAS to CCS trunk interworking (C) • PCM-24/PCM-30 Interoperation (C) • Direct Inward Dialing (C)	• GSCR Sect. 7 • GSCR Sect. 7 • GSCR Sect. 5 • GSCR Sect. 7.1.4 • GSCR Sect. 7.2.2 • GSCR Sect. 4.5.1 • GSCR Sect. 4.5.2 • GSCR Sect. 4.2 • GSCR Sect. 2.5.5 & 2.5.6 • GSCR Sect. 3.10 • GSCR Sect. 7.3 • GSCR Sect. 2.3.2
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Voice	• MOS (R) • MLPP (R) • Secure calls (R)	• CJCSI 6215.01B • GSCR Sect. 3 • CJCSI 6215.01B
		Facsimile	• Analog: TIA/EIA-465-A (R)	• DISR
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	Data	• Modem (VBD) (R) • 56 kbps switched data (R: PRI only) • 64 kbps switched data (R: PRI only) • NX56 synchronous BER (R: PRI only) • NX64 synchronous BER (R: PRI only) • Secure data (STE/STU-III) (R)	• CJCSI 6215.01B • GSCR Sect. 3.10 • GSCR Sect. 3.10 • GSCR Sect. 3.10 • GSCR Sect. 3.10 • CJCSI 6215.01B
		VTC	• ITU-T H.320 (R: PRI only)	• DISR

Table 2. PBX 1 Requirements (continued)

		DSN Line Interfaces		
Interface	**Critical**	**Requirements Required or Conditional**		**References**
2-Wire Analog	Yes	Access	• DN Identification (R) • Line signaling (R) • Loop Start Line (R: 2-Wire Analog only) • Analog Ground Start (R) • Alerting Signals and Tones (R) • WWNDP (R) • Origination Treatment (R) • Termination Treatment (R) • Release Treatment (R) • Interruption Treatment (R) • Connections (R) • Class of Service (C) • 2W user access (R: 2-Wire Analog only) • Analog busy/idle (R: 2-Wire Analog only)	• GSCR Sect. 2.1.1 • GSCR Sect. 5.2 • GSCR Sect. 5.2.1 • GSCR Sect 5.2.2 • GSCR Sect. 5.5 • GSCR Sect. 4.5 • GSCR Sect. 4.1.1 • GSCR Sect. 4.1.2 • GSCR Sect. 4.1.3 • GSCR Sect. 4.1.4 • GSCR Sect. 4.1.5 • GSCR Sect. 4.1.6 • GSCR Sect. 4.3.3 • GSCR Sect. 4.3.4.1
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Voice	• MOS (R) • Announcements (R) • MLPP (R) • Secure Calls (R)	• CJCSI 6215.01B • GSCR Sect. 3.1.3 • GSCR Sect. 3.1, 3.2, 3.2.1, 3.2.2 • CJCSI 6215.01B
2W Digital Proprietary	No	Facsimile	• Analog: TIA/EIA-465-A (R)	• DISR
VoIP (IEEE 802.3)	No	Data	• Modem (VBD) (R) • 56 kbps switched data (R) • 64 kbps switched data (R: BRI only) • NX56 synchronous BER (R: BRI only) • NX64 synchronous BER (R: BRI only) • Secure data (STE/STU-III) (R)	• CJCSI 6215.01B • GSCR Sect. 3.10 • GSCR Sect. 3.10 • GSCR Sect. 3.10 • GSCR Sect. 3.10 • CJCSI 6215.01B
		VTC	• ITU-T H.320 (R: BRI only)	• DISR
		SUT Voice Mail Interfaces		
Interface	**Critical**	**Requirements Required or Conditional**		**References**
2 Wire Digital Proprietary	No	• FCC Part15/Part 68 (R): Analog only • DTMF outpulsing (C) • DISR compliance as applicable (R) • ROUTINE precedence only in accordance with GSCR, Section 3.3 (R) • TIA/EIA-470-B (R): Analog only		• GSCR A7.5 • GSCR A7.5, 5.4.1, 5.4.2 • GSCR A7.5 • GSCR A7.5.5 • GSCR A7.5.1
		Automated Call Distributor Interfaces		
Interface	**Critical**	**Requirements Required or Conditional**		**References**
Internal	No	• DTMF outpulsing (C) • DISR compliance as applicable (R) • ROUTINE precedence only in accordance with GSCR, Section 3.3 (R)		• GSCR Sect. A7.5, 5.4.1, 5.4.2 • GSCR Sect. A7.5 • GSCR Sect. A7.5
		DSN Features & Capabilities		
Feature/ Capability	**Critical**	**Requirements Required or Conditional**		**References**
Common Features	No	• Denied originating service (C) • Code restriction and diversion (C) • Call waiting (C) • Three-way calling (C) • Add-on transfer and conference calling and call hold (C) • Call forwarding (C) • Call pick-up (C)		• GSCR Sect. 2.1.3 • GSCR Sect. 2.1.4 • GSCR Sect. 2.1.5 • GSCR Sect. 2.1.6 • GSCR Sect. 2.1.7 • GSCR Sect. 2.1.8 • GSCR Sect. 2.1.9

Table 2. PBX 1 Requirements (continued)

DSN Features & Capabilities			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
Attendant	No	• Initiate all precedence levels (C) • Visual display (C) • Override class of service (C) • Override busy line (C) • Call deflection (C) • Auto recall (C) • Waiting queue (C)	• GSCR Sect. 2.2.1 • GSCR Sect. 2.2.2 • GSCR Sect. 2.2.3 • GSCR Sect. 2.2.4 • GSCR Sect. 2.2.5 • GSCR Sect. 2.2.6 • GSCR Sect. 2.2.7
Public Safety	No	• Basic Emergency Service (911) (C) • Trace of terminating calls (C) • Outgoing call trace (C) • Trace of a call in progress (C)	• GSCR Sect. 2.4.1 • GSCR Sect. 2.4.2 • GSCR Sect. 2.4.3 • GSCR Sect. 2.4.5
Preset Conferencing	No	• Support 10 bridges; 1 originator and 20 conferees per bridge (C) • Assign up to 20 address numbers per bridge (C) • Use KXX codes for bridge access (C) • Conference notification recorded announcement (C) • Auto retrial and alternate address (C) • Bridge release (C) • Lost connection (C) • Secondary conferencing (C) • Address translation (C)	• GSCR Sect. 2.6 • GSCR Sect. 2.6 • GSCR Sect. 2.6 • GSCR Sect. 2.6.1 • GSCR Sect. 2.6.2 • GSCR Sect. 2.6.3 • GSCR Sect. 2.6.4 • GSCR Sect. 2.6.5 • GSCR Sect. 2.7
Nailed-up Connections	No	• Between any two like terminations (C) • PCM-24 and PCM-30, both CAS and CCS (C) • Supervision passed end-to-end for A/D or D/A (C) • Monitored and auto reconfigure (C) • Support at least 10% of circuits as nailed-up (C) • Non-preemptable (C)	• GSCR Sect. 2.8 • GSCR Sect. 2.8 • GSCR Sect. 2.8 • GSCR Sect. 2.8 • GSCR Sect. 2.8 • GSCR Sect. 2.8
DSN Hotline Services	No	• Hotline restrictions (C) • Auto initiate (C) • Analog and digital (C) • Subscription basis (C) • Protected hotline calling (C) • WWNDP interoperable (C)	• GSCR Sect. 2.12 • GSCR Sect. 2.12 • GSCR Sect. 2.12 • GSCR Sect. 2.12 • GSCR Sect. 2.12.1-4 • GSCR Sect. 2.12.5
ISDN Services	No	• EKTS (C)	• GSCR Sect. 10, Table 10-3
Synchronization	Yes	• Line timing mode (R) • Internal Stratum 4 (R)	• GSCR Sect. 11.1.1.2 • GSCR Sect. 11.1.2.2
Reliability	Yes	• GR-512-CORE (R)	• GSCR Sect. 12
Security	Yes	• GR-815, STIGs, and DIACAP (replacement for DITSCAP) (R)	• GSCR Sect. 13

Table 2. PBX 1 Requirements (continued)

VoIP				
Feature/ Capability	**Critical**	**Requirements Required or Conditional**		**References**
VoIP System	No	VoIP function is conditional. If VoIP is provided, **all** of the following requirements must be met: • Voice Quality with MOS of 4.0 or better • Class of Service (CoS) and Quality of Service (QoS) • ITU-T G.711 PCM Codec • Traffic Engineering • Security • NM • Line timing • Internal Clock • Latency $\leq$ 60 ms • Packet Loss • IPv6 capable		• GSCR App. 3 • GSCR App. 3 • GSCR App. 3 • GSCR App. 3 • GSCR App. 3 • GSCR App. 3 • GSCR App. 3 • GSCR App. 3 • GSCR App. 3 • GSCR App. 3 • GSCR Section 1, paragraph 1.7
Network Gateways				
Gateway	**Critical**	**Requirements Required or Conditional**		**References**
PSTN[1]	No	Trunking	• Positive Identification Control (C) • On-Netting (C) • Off-Netting (C)	• CJCSI 6215.01B • CJCSI 6215.01B • CJCSI 6215.01B
DRSN[2]	Yes	Access	• Alerting Signals and Tones (R) • Call Processing (R) • Call Treatments (R) • Analog busy/idle (R)	• GSCR Sect. 5.5 • GSCR Sect. 4.4 • GSCR Sect. 4.1 • GSCR Sect. 4.3.4.1
		Voice	• MOS (C) • MLPP (C) • Secure calls (C)	• CJCSI 6215.01B • GSCR Sect. 3 • CJCSI 6215.01B

NOTES:
1 Voice, facsimile, data, and VTC service requirements for PSTN are identical to DSN with the exception of MLPP.
2 Facsimile, data, and VTC services are not provided via the DSN to DRSN interface.

JITC Memo, JTE, Extension of the Special Interoperability Test Certification of the Avaya S8400 Digital Switching System with Software Release Communication Manager (CM) 4.0 (R014x.00.2.731.7: Super Patch 14419)

Table 2. PBX 1 Requirements (continued)

LEGEND:					
2W	2-Wire	EKTS	Electronic Key Telephone System	NX64	Data format restricted to multiples of 64 kbps
A/D	Analog to Digital Conversion	EIA	Electronic Industries Alliance	PBX	Private Branch Exchange
ANSI	American National Standards Institute	G.711	Standard for PCM of Voice Frequencies	PCM	Pulse Code Modulation
App.	Appendix	GR	Generic Requirement (Telcordia)	PCM-24	Pulse Code Modulation - 24 Channels
BER	Bit Error Ratio	GR-512	LSSGR: Reliability, Section 12	PCM-30	Pulse Code Modulation - 30 Channels
BRI	Basic Rate Interface	GR-815	Generic Requirements For Network Element/Network System (NE/NS) Security	PRI	Primary Rate Interface
C	Conditional			PSTN	Public Switched Telephone Network
CAS	Channel Associated Signaling	GSCR	Generic Switching Center Requirements		
CCS	Common Channel Signaling	H.320	Standard for Narrowband VTC	Q.735.3	SS7 Signaling Standard for E1 MLPP
		IPv6	Internet Protocol version 6		
CJCS	Chairman of the Joint Chiefs of Staff	ISDN	Integrated Services Digital Network	Q.955.3	ISDN Signaling Standard for E1 MLPP
CJCSI	CJCS Instruction	IT	Information Technology	QoS	Quality of Service
CoS	Class of Service	ITU-T	International Telecommunication Union - Telecommunication Standardization Sector	R	Required
D/A	Digital to Analog Conversion			Sect.	Section
				SS7	Signaling System 7
DIACAP	DoD Information Assurance Certification and Accreditation Process	LAN	Local Area Network	STE	Secure Terminal Equipment
		LSSGR	Local Access and Transport Area (LATA) Switching Systems Generic Requirements	STIGs	Security Technical Implementation Guides
DISR	DoD IT Standards Registry	kbps	kilobits per second	STU-III	Secure Telephone Unit – 3rd Generation
DITSCAP	DoD IT Security Certification and Accreditation Process	KXX	K= any number 2-8; X= any number 1-9	T1	Digital Transmission Link Level 1 (1.544 Mbps)
		Mbps	Megabits per second	T1.619a	SS7 and ISDN Signaling Standard for T1
DN	Directory Number	MFR1	Multi-Frequency Recommendation 1	TIA	Telecommunications Industry Association
DoD	Department of Defense				
DP	Dial Pulse	MLPP	Multi-Level Precedence and Preemption	TIA/EIA-465-A	Group 3 Facsimile Apparatus for Document Transmission
DSN	Defense Switched Network	MOS	Mean Opinion Score		
DRSN	Defense Red Switch Network	ms	milliseconds	VBD	Variable bit data
		NI 1/2	National ISDN Standard 1or 2	VoIP	Voice over Internet Protocol
DTMF	Dual Tone Multi-Frequency	NM	Network Management	VTC	Video Teleconferencing
		NX56	Data format restricted to multiples of 56 kbps	WWNDP	Worldwide Numbering and Dialing Plan
E1	European Basic Multiplex Rate (2.048 Mbps)				

Table 3. SUT PBX 1/DVX Comparison and Interoperability Test Summary

GSCR Paragraph	Requirement	PBX 1 Critical	DVX Critical	Status	Remarks
2.1.4	Code Restriction and Diversion	No	Yes	Certified	Met all critical CRs and FRs.
2.4.2	Trace of Terminating Calls	No	Yes	Certified	Met all critical CRs and FRs.
2.4.3	Outgoing Call Trace	No	Yes	Certified	Met all critical CRs and FRs.
2.4.4	Tandem Call Trace	No	Yes	Certified	Met all critical CRs and FRs.
2.4.5	Trace of a Call in Progress	No	Yes	Certified	Met all critical CRs and FRs.
2.5.4.2	Manual Test of Trunks	No	Yes	Certified	Met all critical CRs and FRs.
2.5.5	Trunk Group Remove from Service (Make Busy)	No	Yes	Certified	Met all critical CRs and FRs.
2.5.6	Trunk Group Return to Service (Make Idle)	No	Yes	Certified	Met all critical CRs and FRs.
2.5.7	Carrier Group Alarm	No	Yes	Certified	Met all critical CRs and FRs.
A2.5.2.1	Preset Conferencing	No	Yes	Certified	Met all critical CRs and FRs.
2.12.1	Protected Hotline Calling	No	Yes	Certified	Met all critical CRs and FRs.
2.12.2	Hotline Service Protection	No	Yes	Certified	Met all critical CRs and FRs.

JITC Memo, JTE, Extension of the Special Interoperability Test Certification of the Avaya S8400 Digital Switching System with Software Release Communication Manager (CM) 4.0 (R014x.00.2.731.7: Super Patch 14419)

Table 3. SUT PBX 1/DVX Comparison and Interoperability Test Summary (continued)

GSCR Paragraph	Requirement	PBX 1 Critical	DVX Critical	Status	Remarks
2.12.3	Non-Pair Protected Hotline Calling	No	Yes	Certified	Met all critical CRs and FRs.
2.12.4	Pair Protected Hotline Calling	No	Yes	Certified	Met all critical CRs and FRs.
3.2.3	MLPP Trunk Selection	No	Yes	Certified	Met all critical CRs and FRs.
3.2.4.1	Calls from non-MLPP Networks	No	Yes	Certified	Met all critical CRs and FRs.
3.2.4.2	Precedence Calls to non-MLPP	No	Yes	Certified	Met all critical CRs and FRs.
3.4.1	Channel Associated Signaling	No	Yes	Certified	Met all critical CRs and FRs.
3.7	ISDN MLPP Primary Rate Interface (ANSI T1.619a)	Yes	No	Certified	Met all critical CRs and FRs.
3.14	Data Collection	No	Yes	Certified	Met all critical CRs and FRs.
4.1.6	Class of Service	No	Yes	Certified	Met all critical CRs and FRs.
4.2	Primary and Alternate Routing	No	Yes	Certified	Met all critical CRs and FRs.
4.3.1	E&M Lead Signaling States	No	Yes	Certified	Met all critical CRs and FRs.
4.3.2	Four Wire E&M Analog User Access Lines	No	Yes	Certified	Met all critical CRs and FRs.
4.4.2	Terminating Call Processing	No	Yes	Certified	Met all critical CRs and FRs.
4.5.2	DSN Switch MFR1 Outpulsing Digit Format	No	Yes	Certified	Met all critical CRs and FRs.
4.5.1.8	Emergency Service 911 Conflict Resolution	Yes	No	Certified	Met all critical CRs and FRs.
Table 4-9	DSN Switch MFR1 Outpulsing Digit Format	No	Yes	Certified	Met all critical CRs and FRs.
Table 4-10	DSN Switch DTMF Outpulsing Digit Format	No	Yes	Certified	Met all critical CRs and FRs.
4.5.5	Base Services – Abbreviated Numbers	No	Yes	Certified	Met all critical CRs and FRs.
4.5.7	Digit Registration Capacity	No	Yes	Certified	Met all critical CRs and FRs.
4.5.8	Screening	No	Yes	Certified	Met all critical CRs and FRs.
5.3.3.1.1	Wink Start	No	Yes	Certified	Met all critical CRs and FRs.
5.3.3.1.2	Glare Operation	No	Yes	Certified	Met all critical CRs and FRs.
5.3.3.2.1	Wink Start	No	Yes	Certified	Met all critical CRs and FRs.
5.3.3.2.2	Glare Resolution	No	Yes	Certified	Met all critical CRs and FRs.
5.3.7	Satellite Interface	No	Yes	Certified	Met all critical CRs and FRs.
5.3.8	Disconnect Control	No	Yes	Certified	Met all critical CRs and FRs.
5.3.9	Reselect or Retrial	No	Yes	Certified	Met all critical CRs and FRs.
5.3.10	Off-Hook Supervision	No	Yes	Certified	Met all critical CRs and FRs.
5.4.1	Dial Pulse Signals	No	Yes	Certified	Met all critical CRs and FRs.
5.4.2	MFR1 Signaling	No	Yes	Certified	Met all critical CRs and FRs.
5.4.3	MFR1 2/6 Signaling	No	Yes	Certified	Met all critical CRs and FRs.
7.1.2	Supervisory Channel Associated Signaling	No	Yes	Certified	Met all critical CRs and FRs.
7.2	PCM-30 Digital Trunk Interface	No	Yes	Certified	Met all critical CRs and FRs.
7.3	Interoperation of PCM-24 and PCM-30 Systems	No	Yes	Certified	Met all critical CRs and FRs.
A2.5.2.5	DISA Network Traffic Management Operating System (NTMOS)	No	Yes	Certified	Met all critical CRs and FRs.
A2.5.2.6	Data Quality	No	Yes	Certified	Met all critical CRs and FRs.
9.2.2.1.1	Traffic Measurements	No	Yes	Certified	Met all critical CRs and FRs.
9.8	Remote Access to Switch	No	Yes	Certified	Met all critical CRs and FRs.
A2.5.2.3	DVX Switch ISDN Outpulsing Digit Formats	No	Yes	Certified	Met all critical CRs and FRs.
12.2	PBX Availability	Yes	No	Certified	Met all critical CRs and FRs.
Section 13	Security	Yes	No	Certified	Met all critical CRs and FRs.

NOTE: The requirements for PBX 1s and DVXs are identical except for those listed in above.

JITC Memo, JTE, Extension of the Special Interoperability Test Certification of the Avaya S8400 Digital Switching System with Software Release Communication Manager (CM) 4.0 (R014x.00.2.731.7: Super Patch 14419)

Table 3. SUT PBX 1/DVX Comparison and Interoperability Test Summary (continued)

LEGEND:			
A	Appendix	GSCR	Generic Switching Center Requirements
ANSI	American National Standards Institute	ISDN	Integrated Services Digital Network
BRI	Basic Rate Interface	MFR1	Multi-Frequency Recommendation 1
CDR	Call Detail Recording	MLPP	Multi-Level Precedence and Preemption
CRs	Capability Requirements	NI 1/2	National ISDN Standard 1 or 2
DISA	Defense Information Systems Agency	PBX	Private Branch Exchange
DSN	Defense Switched Network	PCM-24	Pulse Code Modulation - 24 Channels
DSS1	Digital Subscriber Signaling 1	PCM-30	Pulse Code Modulation - 30 Channels
DTMF	Dual Tone Multi-Frequency	SS7	Signaling System 7
DVX	Deployable Voice Exchange	SUT	System Under Test
E&M	Ear and Mouth	T1	Digital Transmission Link Level 1 (1.544 Mbps)
FRs	Feature Requirements	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1

5. No detailed test report was developed in accordance with the Program Manager's request. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Unclassified-But-Sensitive Internet Protocol Router Network (NIPRNet) e-mail. More comprehensive interoperability status information is available via the JITC System Tracking Program (STP). The STP is accessible by .mil/gov users on the NIPRNet at https://stp.fhu.disa.mil. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at http://jit.fhu.disa.mil (NIPRNet), or http://199.208.204.125 (SIPRNet). Information related to DSN testing is on the Telecom Switched Services Interoperability (TSSI) website at http://jitc.fhu.disa.mil/tssi.

6. The JITC point of contact is Mr. Joseph Roby, DSN 879-0507, commercial (520) 538-0507, FAX DSN 879-4347, or e-mail to joseph.roby@disa.mil. The JITC's mailing address is P.O. Box 12798, Fort Huachuca, AZ 85670-2798. The tracking number for the SUT is 0700902.

FOR THE COMMANDER:

Enclosure a/s

for RICHARD A. MEADOR
Chief
Battlespace Communications Portfolio

JITC Memo, JTE, Extension of the Special Interoperability Test Certification of the Avaya S8400 Digital Switching System with Software Release Communication Manager (CM) 4.0 (R014x.00.2.731.7: Super Patch 14419)

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Defense Information Systems Agency, GS23

ADDITIONAL REFERENCES

(c)   Chairman of the Joint Chiefs of Staff Instruction (CJCSI) 6215.01B, "Policy for Department of Defense Voice Services," 23 September 2001

(d)   Joint Interoperability Test Command, Memo, JTE, "Special Interoperability Test Certification of the Avaya S8400 Digital Switching System with Software Release Communication Manager (CM) 4.0 (R014x.00.2.731.7: Super Patch 14419)," 30 October 2007

(e)   Defense Information Systems Agency, "Department of Defense Voice Networks Generic Switching Center Requirements (GSCR), Errata Change 2," 14 December 2006, Revised 27 March 2007

(f)   Joint Interoperability Test Command, "Defense Switched Network Generic Switch Test Plan (GSTP), Change 2," 2 October 2006

Enclosure